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SIRIUS-XM JOINT OPPOSITION
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A TECHNICAL REPORT REGARDING CODING EFFICIENCY AND THE SIRIUS-XM MERGER

A Technical Report Regarding Coding Efficiency and the *Sirius-XM* Merger

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July 24, 2007

1 Introduction

In a March 16, 2007, report entitled "An Engineering Statement: Prepared on Behalf of the National Association of Broadcasters"[1] the author presents an analysis of modem and compression technologies used in the Sirius and XM satellite radio systems and concludes there is no opportunity for providing "best of" programming from each system on the other system without unacceptable degradation in audio quality or without jettisoning existing channels. To the contrary, from a technical point of view, a merger of the satellite companies would allow expanded programming offerings that benefit substantially existing and future customers of Sirius and XM.

Specifically, both of the satellite radio providers have independently developed a rich suite of transmit side technologies the consolidation of which would yield improved coding efficiency for each system without requiring any changes to the communications infrastructure and/or currently deployed receivers. Transmission bandwidth freed up through such increases in coding efficiency could be used to offer some of the most compelling content from one service to the other and vice versa (*i.e.*, "best of" programming) without sacrificing audio quality on either system.

The following sections of this report describe the parameters and the data that lead to the above conclusion and highlight several fundamental problems we see in the analysis presented in [1].

2 Technology Consolidation and the Resulting Improved Coding Efficiency

The assumptions that (i) audio compression codecs are in some sense frozen in time; and (ii) the only way to achieve higher channel efficiency (*i.e.*, lower per channel average bit rate) is by sacrificing audio quality stand out as fundamental flaws in the analysis in [1]. While this may have been the case with speech codecs utilized in earlier generations of satellite communication systems [2], this is far from the case for today's state-of-the-art audio codecs.

Most -- if not all -- modern audio compression techniques are based on the "perceptual coding paradigm" whereby substantial emphasis is placed on accurately modeling the human hearing system. Because our understanding of natural phenomena such as human hearing is not perfect, audio codecs are generally designed so that substantial improvement in audio quality is possible as scientific research in human perception continues and more accurate models for human hearing become available. Thus, the architecture of audio codecs is perhaps best viewed as a collection of promising coding tools that can be configured and combined by software using a description language or *bitstream* assembled at the encoder end and transmitted to the decoder at the receiver end. It is therefore a dynamic framework for continued innovation. This is true for both proprietary[3][4] and standards-based[5][6][7][8] audio compression technologies. The practical import of this observation is that encoders have a fairly long evolutionary cycle even after the basic architecture of the codec is finalized. The analysis presented in [1] completely overlooks this aspect of audio coding as well as the role of encoder improvements in audio signal compression.

Section 2.1 below examines three case studies illustrating the impact of encoder upgrades on real-life systems. Section 2.2 presents data that explains why there is a strong reason to believe that Sirius and XM have access to complementary technology pieces, such that the combination of the companies will accelerate the process of audio quality improvement through encoder upgrades. This will in turn free up substantial capacity on both services in the near future (following merger approval). Expanded program offerings will, therefore, be possible by allowing the best content of both services to be available to all U.S. satellite radio subscribers.

2.1 Real Life Scenarios Illustrate the Impact of Encoder Upgrades

Case Study 1: The Evolution of MP3 Encoders

The evolution of the wildly popular MP3 codec (or, more precisely, the ISO/IEC MPEG-1 Audio Layer III codec) offers a fascinating illustration of the scope and power of encoder upgrades in modern audio codecs. The basic structure of MP3 was finalized as part of the standard ISO/IEC 11172-3, published in 1993[8]. Further work on this standard, including lower bit rate extensions, occurred in 1994 as part of MPEG-2 (ISO/IEC 13818-3)[9]. The standard specification did not actually explain how to build an encoder. Therefore, soon after the standard was finalized, numerous MP3 encoders with widely varying levels of audio quality became available. Interestingly, even the best encoder available at that time exhibited audio quality somewhere between annoying and barely acceptable at 128 kbps. By 1998, MP3 at 128kbps was still providing quality only at the level of AAC-LC at 96 kbps (a newly standardized MPEG codec at that time, which was still early in its own encoder evolution cycle)[10].

Summary data from the test results in [10] are extracted in Figure 1 below. In this figure, the x axis shows the codec under test and the y axis indicates its overall rating expressed as Mean Opinion Score (MOS) on the *Diff Impairment Scale* using the ITU-R BS1116 subjective audio testing methodology [11] (also known as the triple stimulus/hidden reference/double blind methodology. In this test the codec quality is measured on a 5 point MOS scale: 5.0 = Imperceptible Distortion, 4.0 = Perceptible but not annoying, 3.0 = Slightly annoying, 2.0 = Annoying, 1.0 = Very annoying. The final score is typically presented on a Difference Scale, Diff Score = 5.0 – MOS Score, so that a *negative value of the Diff Score that is closer to zero implies higher quality*) In Figure 1, the solid square dot indicates the actual *Diff Score* and the range above and below the dot indicates the 95% confidence interval associated with the score. It is obvious from this figure that in this study the relatively mature MP3 codec at 128 kbps of 1998 (provided by the industry leader in this area at the time Fraunhofer IIS - Audio and Multimedia Group: http://www.iis.fraunhofer.de/bf/amm/) was found to be statistically inferior to the new 1998 codec AAC LC at 128 kbs (the highest edged of the MP3 128 score 95% confidence interval is lower than the lower edge of AAC LC 128 score 95% confidence interval).

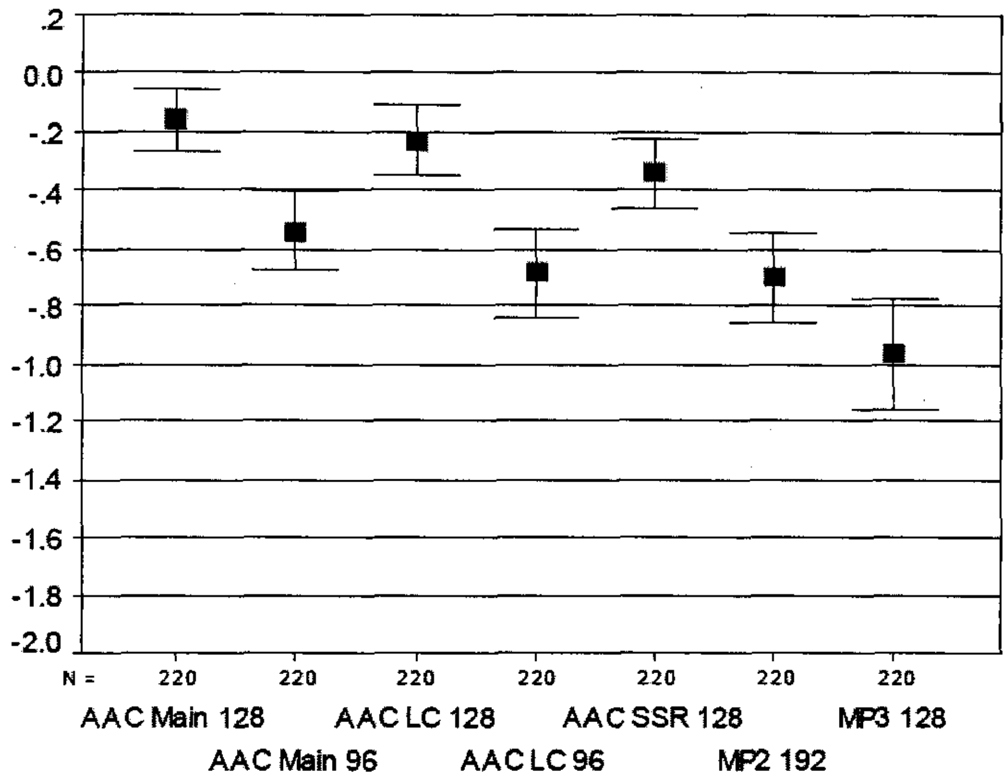
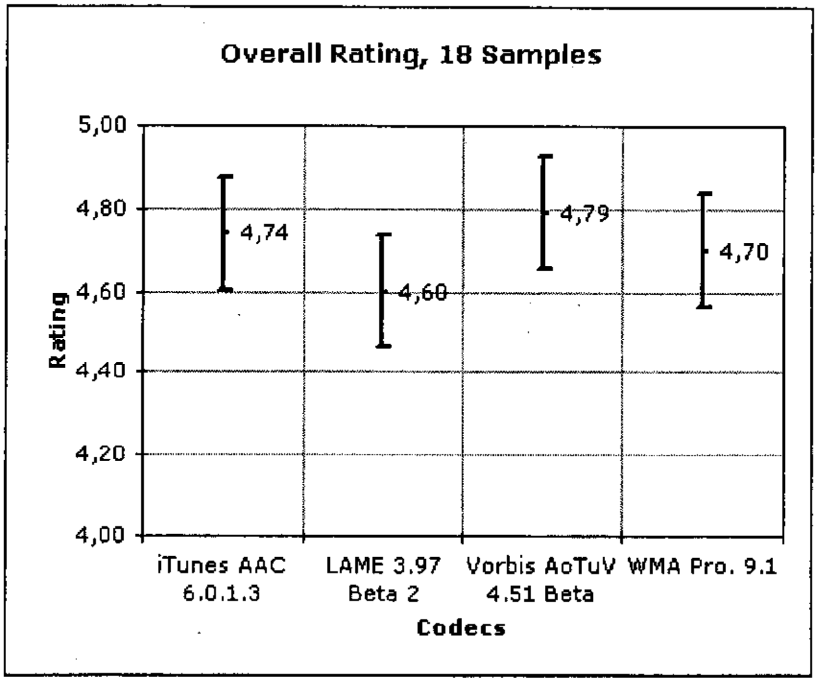


Figure 1: Summary test results from a comparison of audio quality in different codecs at 128 kbps extracted from [11], 1998.

However, recent testing with the latest MP3 encoders (LAME 3.97 Beta 2 encoder in this specific study) indicates that the quality at 128 kbps has improved substantially and is statistically indistinguishable from the latest ACC encoders (*e.g.*, Apple iTunes AAC 6.0.1.3) at 128 kbps [12].

A summary of the results in [12] is provided below. Comparing the results in Figure 2 with those in Figure 1, it becomes clear that although in 1998 the best of MP3 encoders were statistically inferior to even the newly standardized AAC encoders, by 2006, the situation was quite different. In 2006, the latest MP3 codecs, *purely through encoder* *enhancements*, were able to provide audio quality at a level that was statistically indistinguishable from the latest AAC codec. (In Figure 2 the scores are expressed directly on the MOS scale so higher score reflects better quality. The 95% confidence intervals of MP3 codec – LAME 3.97 Beta 2 - and AAC codec – iTunes AAC 6.0.1.3 - overlap substantially indicating statistical equivalence.)



Full results are available on
http://www.maresweb.de/mf-128-1-results

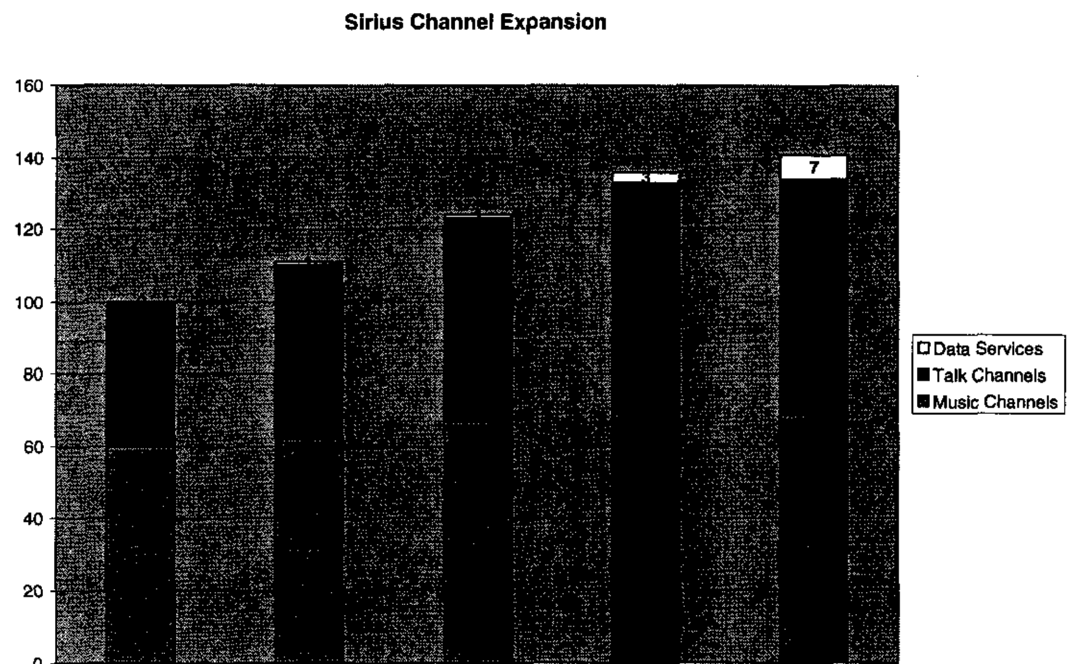
Figure 2: Summary test results from a comparison of audio quality in different codecs at 128 kbps extracted from [12], 2006.

Moreover, MP3 coded music at 64 kbps (or even lower) now circulates widely and exhibits acceptable quality in many applications.

Case Study 2: Increased Bandwidth Efficiency Achieved by Sirius Satellite Radio System since Service Launch

The Sirius satellite radio system itself provides another compelling demonstration of the power and potential of encoder upgrades. In particular, the evolution of the capacity of the Sirius system shows that incorporating improved technology at the encoder can result in far more efficient encoding while maintaining audio quality.

Sirius first launched its service in early 2002 with a total of 100 music and news channels. Shortly thereafter, through transmitter side upgrades, Sirius added about 25 additional programming channels. The figure below illustrates this dramatic increase in the number of channels offered by Sirius over the course of its 5 years of operation and shows the efficiency realized through improvements on the encoding side of the broadcasting chain. These include not only additional programming channels but also data channels to provide listeners with weather forecasts as well as traffic reports, movie information, sports scores, and fuel prices.



	2002	2003	2004	2005	2006
Data Services	0	1	1	3	7
Talk Channels	40	48	56	64	65
Music Channels	60	62	67	69	69

Figure 3

It is important to note that all of the deployed receivers continue to function notwithstanding the growth in the number of channels to well above the maximum number posed in [1]. In other words the addition of new channels was achieved purely through encoder improvements and without impairing the functioning of the earlier receivers.

Case Study 3: Increased Bandwidth Efficiency Achieved by WorldSpace Satellite Radio System since Service Launch

The evolution of WorldSpace Satellite Radio Service (http://www.worldspace.com) which offers digital satellite radio to people around the world through its two satellites is a similar study in how increased bandwidth efficiency can be achieved through transmit side improvements over time. For example, during its initial phase of operations, WorldSpace offered only about 30 channels of programming to its Asia subscribers. In 2005, it announced the addition of about 19 new programming channels[24] and offers over 50 channels to its Asia subscribers at present. To the best of our knowledge this increase in programming choices by Worldspace has also been achieved without making any of its older *receivers obsolete.*

2.2 Technology Tools Available to Sirius and XM for Improved Encoder Efficiency

Below we describe three sets of tools with proven ability to achieve higher coding efficiency. Independently, Sirius and XM have committed substantial resources in attempting to develop these tools. However, the efforts have progressed along separate tracks. While Sirius has focused on improving coding efficiency through improved Psychoacoustic Modeling and Joined Encoding (i.e. *statistical multiplexing*)[13], XM has focused more on optimizing the pre-processor configuration[14], particularly in optimizing the overall interaction between the pre-processor and the audio encoder.

Combining these efforts likely would yield improved efficiency on both the systems. The 3 available tools for realizing improved efficiency through transmit side upgrades are therefore: (i) Improved Psychoacoustic Modeling (ii) Statistical Multiplexing, and, (iii) Optimized Pre-processing. These tools are further described and analyzed below.

A. Psychoacoustic Modeling

Psychoacoustic modeling offers significant quality improvement and has substantially reduced the bit rate requirements for transmitting good quality audio signals. The professional literature is replete with examples of such bit rate reductions.[15][16] More specifically, several new results and enhancements in the area of Psychoacoustic Modeling have significantly improved audio quality; examples are codecs such as AAC and PAC[16][17][18] (as reported in [1], the XM system uses an extension of the AAC[6] codec similar to the standard described in [19], whereas the Sirius system uses PAC[3] as the audio compression scheme in their respective systems)

The data presented below show how introducing an *Enhanced Psychoacoustic Model* (*EPM*) developed for Sirius and used in PAC can also improve the coding efficiency of the AAC codec. To illustrate this point, the default Psychoacoustic model used in AAC [19][20] was replaced by a version of *EPM*. The average *per frame bit demand* used to encode audio with perceptually similar audio quality was then measured using a database of critical audio material.[1,2] Figure 4(a) below (*i.e.*, the "before" diagram) shows the distribution (histogram) of per frame bit demand for the AAC codec with its standard

[1] A "frame" is collection of consecutive audio samples which is used as a unit of analysis and coding by audio codecs. Both PAC and AAC typically employ a frame size of 1024 audio samples. Per frame bit demand refers to the numbers of bits required to encode a single frame of audio.

[2] Audio quality may be measured using Subjective [11] and Objective [26] Measures. In this study we ensured that the audio quality is identical or better with the *Enhanced Psychoacoustic Model (EPM)* using a combination of both.

Psychoacoustic model. Figure 4(b) (*i.e.*, the "after" diagram) shows similar distribution (histogram) of per frame bit demand for the AAC codec with the *EPM. The distribution* of gains (*i.e.*, reduction in bit demand due to *EPM*) is summarized in Figure 4(c). This illustrates the reduction in bit demand for identical (or very similar) subjective and objective audio quality. In other words Figure 4 (c) is a histogram of bits saved (effectively, a gain) for each frame through Psychoacoustic Model changes which increase coding efficiency without degrading audio quality (i.e., introduction of *EPM* in AAC). All the histograms in Figures 4 (a), (b), (c) are plotted using a large audio database of about 8000 frames.

On average, a bit demand reduction of about 8.5% was observed in our test database for a target bit rate of 51 kbps. It should be noted that this was a *first level* integration of EPM in the AAC codec without extensive joint optimization of the interaction of the Psychoacoustic Model with some of the other coding tools in AAC. Therefore, additional gains should be realized with further optimizations. *It may further be noted that the audio quality as measured by subjective and objective analysis was actually noticeably better using the EPM for stereo audio signals.*

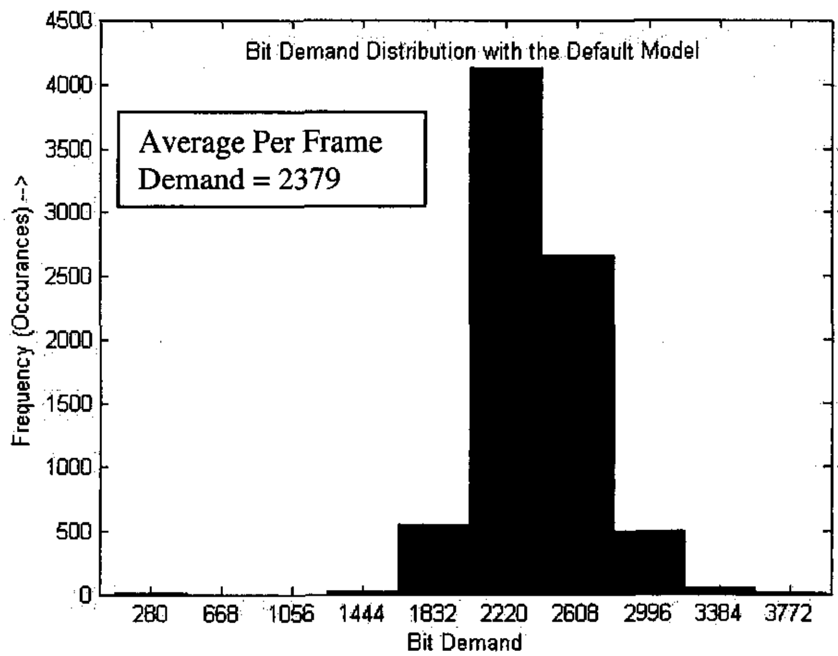


Figure 4(a): Distribution of per frame bit demand in the AAC codec using the standard Psychoacoustic Model.

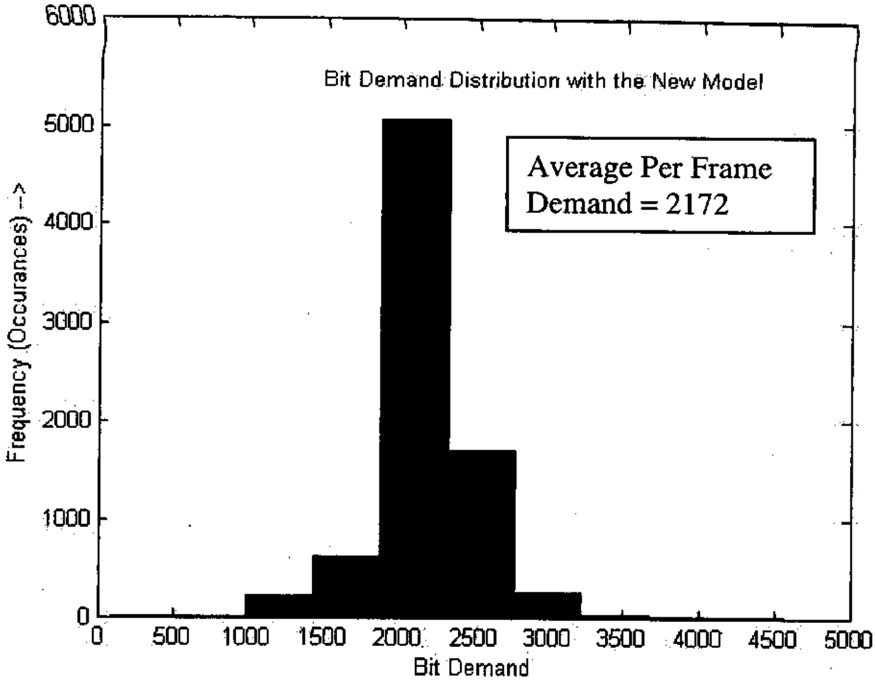


Figure 4(b): Distribution of per frame bit demand in the AAC codec using the Enhanced Psychoacoustic Model (EPM).

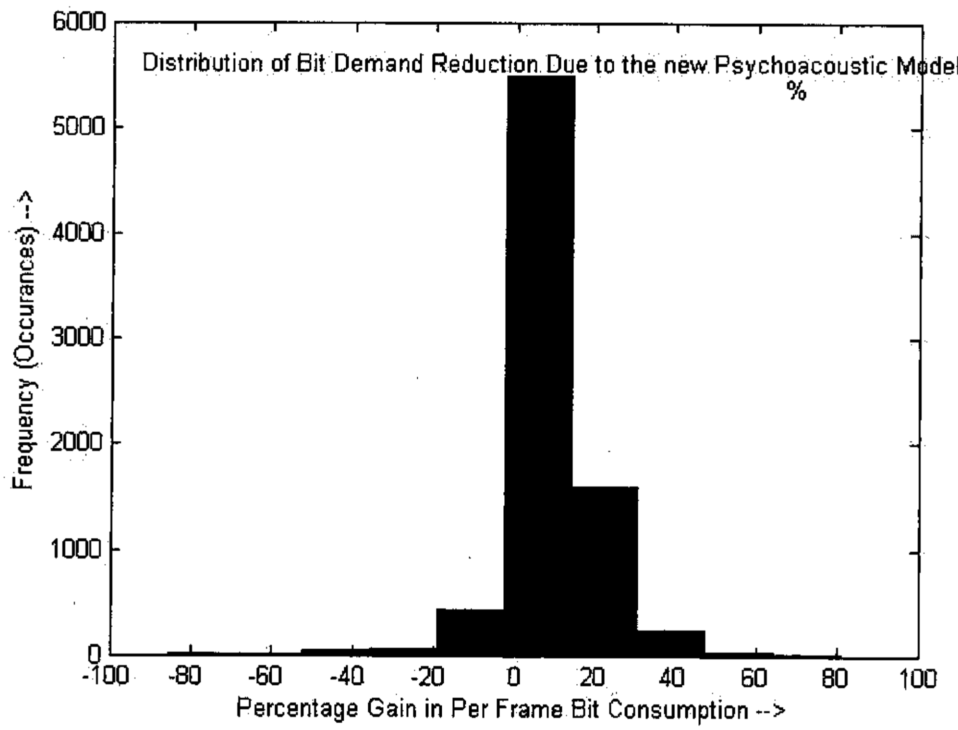


Figure 4(c): Distribution of gains (reduction) in frame bit demand in the AAC codec using the *Enhanced Psychoacoustic Model* (*EPM*).

To summarize, Figure 4(a) above shows an average bit demand of 2379 bits/frame when *using the default Psychoacoustic Model in AAC.* The average bit demand goes down to 2172 bits once *EPM* is integrated into AAC. Notably, the audio quality remains the same (or was found to be even improved for stereo audio signals). The resulting improvement in coding efficiency is summarized in Figure 4 (c) which shows a distribution of per frame bit gain (expressed as a percentage) through the use of EPM. On an average a gain of 8.5% was achieved over a database of critical audio test samples.

B. Statistical Multiplexing

In this section we assess the manner in which statistical multiplexing can be used to achieve higher compression efficiency [21]. Statistical multiplexing operates by encoding multiple channels instead of a single channel such that channel capacity can be adaptively steered from one channel to the other. This improves channel utilization by allowing the bandwidth to be divided based on need. This technique is already in use in the Sirius system and it is our understanding that the XM transmission system accommodates the necessary "hooks" to invoke similar features. Additional processing power, including advanced distributed computing infrastructure, is needed to fully realize the potential gains (in compression efficiency) promised by statistical multiplexing. Therefore, as more powerful computing environments and networks become available, incorporation of such gains on the transmit side, along with better statistical multiplexing algorithms, will reduce average bit rate requirements while maintaining or improving quality.

Figure 5 below illustrates how statistical multiplexing is a simple yet powerful concept for utilization in a broadcast system like that used by XM and Sirius where multiple audio channels share the same available bandwidth. The individual curves in this Figure represent the Cumulative Density/Distribution Function (CDF)[22] for the distribution of per frame bit demand (for an explanation of per frame bit demand see the section above on Psychoacoustic Modeling) for a particular audio channel for distortion-free audio encoding.

Audio signals are, in general, highly non-stationary and the complexity of a program channel can vary substantially from frame to frame resulting in wide variations in the per frame bit demand over time. Therefore, in general, there may be a significant loss in quality at constant bit rate coding as the audio codec attempts to fit a frame with high bit demand into an allocated bandwidth that cannot support the frame. Audio codecs attempt to minimize this quality loss by using techniques such as bit allocation buffer[3][5]. Such variation is of course quite pronounced for speech type signals with periods of silence but is actually found for all type of audio including rock, classical, vocal, instrumental, etc. The randomness of the individual curves in Figure 5 illustrates the difficult task encountered by the broadcast system manager in deciding what capacity or bit rate should be allocated to a particular audio channel to ensure consistent audio quality at all times. It is obvious that a level of sub-optimality in bit rate allocation is almost guaranteed as one is forced to take a somewhat conservative approach to ensure acceptable audio quality.

In Figure 5, the bold smooth curve (marked with * symbols) is the CDF of the average bit demand for the individual channels multiplexed together. The smoothness of this curve illustrates how a fixed capacity can be easily allocated to this *cluster* of audio channels while maintaining consistent audio quality if they are jointly encoded. Figure 5 also shows a measure of statistical multiplexing gain as computed in [21] which for this particular cluster of randomly chosen audio signals was measured at 13 dB. As noted above, the behavior like the one demonstrated in Figure 5 is noticed for all types of audio materials; as long as about 10 or more audio channels are multiplexed together substantial statistical multiplexing gain is achievable.

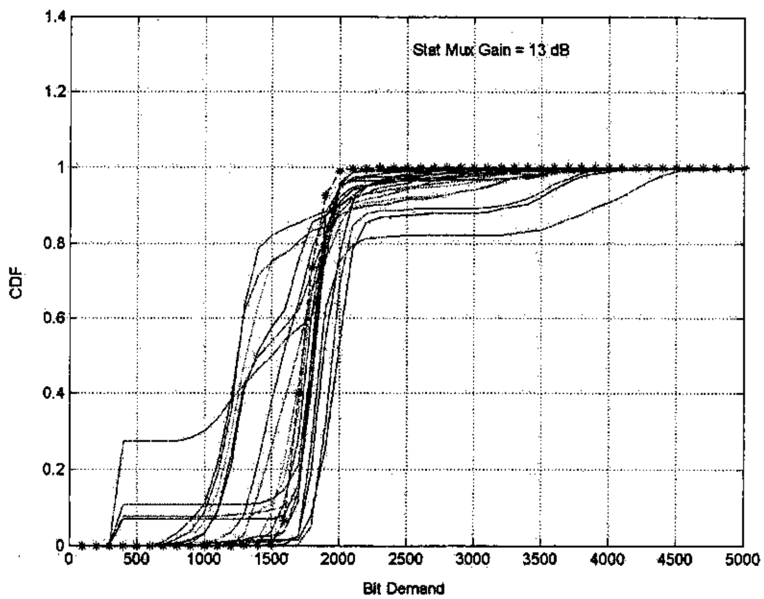


Figure 5: Cumulative density functions for joint encoding and individual encoding for randomly chosen audio channels. The smoothness of *average* curve (*) illustrates the statistical multiplexing gain.

Statistical multiplexing is a proven technology in the area of video broadcasting and is used widely to achieve up to 30% to 40% efficiency on a transmission channel. It has been reported that even the latest video codecs such as H.264[25] benefit substantially through statistical multiplexing techniques[26]. Similar to the audio case discussed above, in the video codec case if an encoder operates in a constant bit rate mode, then complex scenes and picture changes do not get enough bandwidth and artifacts are observed. Therefore, one is forced to allocate a high bandwidth (bit rate) for acceptable quality which is then wasteful in the case of simple scenes that would not require as much bandwidth. Moreover, in a set of multiple video channels the chances of complex scenes appearing on more than one channel is relatively rare. Statistical multiplexing takes advantage of these realities in order to support additional channels.

C. Audio Preprocessor Customized for Codecs

This has been a key area of focus for XM, which currently incorporates Neural Audio pre-processing software that uses algorithms optimized to correct distortions introduced by codecs[27]. Such optimization of temporal and spectral elements prior to encoding further improves sound clarity and intelligibility. The benefits of such pre-processing apply to all codecs.

To illustrate the cumulative benefit of the above coding tools (Psychoacoustic Model, Statistical Multiplexing, Customized Preprocessor), we reproduce the data presented in Table 2 of [1] and estimate the numbers of additional channels which may be added to the XM service assuming a conservative 8.5% increasing in coding efficiency (the gain achievable by Psychoacoustic Model improvements alone. (As described above, additional efficiency gains are possible, *e.g.*, through the use of statistical multiplexing).

Table 1

Program channel genre/type	**Average bit rate/channel (kbps)**	**Number of Program Channels (Current)**	**Estimated Number of New Channels which may be added (@8.5% coding efficiency gain)**
Rock Music	40	31	3
Hip-Hop/Rap Music	40	13	1
Classical Music	50	7	1
Jazz Music	40	15	1
Country Music	40	10	1
Voice/News/Comedy	16	50	4
Weather/Traffic	4	22	-
Program Data	8	1	-
	TOTAL	148	11

2.3 The Inapplicability of NPR Subjective Test to the Situation Concerning Sirius and XM

In [1] an attempt was made to project the results of a subjective audio quality test conducted for NPR in 2004 ("NPR study"[28]) to the situation regarding Sirius and XM. We believe that this is a flawed comparison in that

1. the NPR study is only a snapshot in time (*i.e.*, Oct. 2004) and not only have there been improvements in codec technologies since the study, but also use of this test data to the present situation completely ignores the proven track record of audio codecs to provide substantially improved quality over time purely through encoder optimizations;

2. In the codecs used in NPR study no attempt was made to optimize the compression algorithm at the specified bit rates. The NPR study was a "one size fits all" approach whereas satellite radio providers use advanced coding techniques, such as enhanced psychoacoustic modeling, pre-processing optimization, and statistical multiplexing, to optimize each bit rate of interest independently;
3. In terrestrial digital audio broadcasting, codecs have been tailored in a special way to address specific issues pertinent to these systems such as the issue of how the digital signal blends to analog, and the need to provide layered (core plus enhancement) type coding. These factors are not present in satellite radio service removing some of the constraints which may inhibit the optimization of audio codecs; and
4. Terrestrial digital audio broadcasting does not present (or at least does not use) the same opportunities for statistical multiplexing as Satellite radio. Satellite radio providers transmit dozens of channels in a contiguous swath of spectrum and the receiver is a "broad-band" tuner that receives all these channels together, which, as explained above, offers great opportunities for statistical multiplexing and the associated coding efficiency optimization.

3 Conclusion

In sum the report set forth in [1] simply fails to take into account the substantial improvements in system performance that have been – and can still be – achieved through upgrades in the encoding of audio programming without changes to the current generation of receivers. As noted herein, these improvements can result in additional channels of programming being made available on existing satellite systems without diminishing the audio quality enjoyed by listeners and without eliminating current channels. As such, opportunities for improving the services available to both XM and Sirius subscribers can flow from the merger as each system benefits from the development work of the other which has been of complementary nature to date.

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BIOGRAPHY OF DR. DEEPEN SINHA

Dr. Sinha has been focused in the area of audio compression technology for over a decade. He serves ATC Labs as the President. He has contributed to numerous technical publications in the area and is an inventor of over 25 awarded or applied patents on audio coding and joint source-channel coding. He started his career in AT&T Bell Labs research in 1993, where he contributed to the early versions of AT&T Perceptual Audio Coder (PAC) and developed a multi-channel version of PAC. In 1995, after the split of AT&T, Dr. Sinha transitioned to Lucent Technologies Bell Labs and between 1995 and 1998, was the principal researcher for Lucent in the Audio Signal Compression area. His work there formed the basis for the latest generation of Lucent's PAC Codec.

Between 1998 and 2003, he worked at Lucent Digital Radio and its successor iBiquity Digital, and during this time focused on 3 main areas: (i) improving PAC audio quality at bit rates below 64 kbps, (ii) Statistical Multiplexing for multi-program audio broadcasting, and, (iii) Multistreaming for robust broadcast quality under channel impairments, *e.g.*, interference. During this time he led an integrated team of 10 research and development engineers. A primary achievement of this team was the development and integration of the PAC4 codec (an upgrade to Lucent PAC), and the statistical multiplexing technology into the Sirius Satellite Radio system.

Since the formation of ATC Labs, Dr. Sinha has focused on developing novel techniques for exploiting signal redundancies to achieve high audio quality at very low bit rates. He has also worked on developing a deeper understanding of Perceptual Modeling and quantization issues in audio codecs related to certain type of signals which present unique challenges such as vocals and some tonal instruments.

Deepen Sinha earned a B.Tech (Hons.) degree in Electronics & Electrical Communications Engineering from the Indian Institute of Technology, Kharagpur, India, in 1986, a M.S. degree in Electrical Engineering from Iowa State University in 1989, and a Ph.D. in Electrical Engineering from University of Minnesota, Minneapolis, in 1993. His Ph.D. dissertation made original contributions to the idea of wavelet based audio coding.

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EXHIBIT H

SIRIUS-XM JOINT OPPOSITION
MB Docket No. 07-57
July 24, 2007

NEURAL AUDIO CORPORATION,
REPORT REGARDING CERTAIN TECHNICAL
ASPECTS OF THE SIRIUS-XM MERGER

neural audio.

Neural Audio Corporation
Report Regarding Certain Technical Aspects
Of the XM-Sirius Merger

July 24, 2007

1. Introduction and Summary

In connection with the proposed merger of XM and Sirius, Neural Audio Corporation ("Neural Audio") has been asked to review a document prepared by Meintel, Sgrignoli & Wallace entitled "An Engineering Statement Prepared on Behalf of the National Association of Broadcasters Regarding the Technical Aspects of the SDARS Providers XM and Sirius" (March 16, 2007) ("NAB Engineering Statement"). The NAB Engineering Statement concludes (at page 9) that XM and Sirius "are limited in their ability to add new program channels to their services without removing an equivalent number of existing program channels," and that "[a]ttempts to achieve more program capacity through more aggressive digital compression and fewer bits per program would result in significant audio quality degradations likely to be unacceptable to consumers."

As we explain below, these conclusions are at odds with well-established technological trends in the broadcasting industry. Continuous advancements in audio compression technologies have allowed many audio entertainment providers—including broadcasters as well as both of the satellite radio providers—to provide more programming over the same amount of bandwidth without any corresponding degradation in audio quality, even with reductions in bit rates. Neural Audio has contributed to that progress by developing and improving solutions that XM has employed successfully to substantially increase the number of channels it offers. We describe this broader technological trend and Neural Audio's product below. In addition, we address certain assertions contained in the NAB Engineering Statement regarding the satellite radio providers' ability to add programming to their existing channel line-ups.

2. Innovations in Audio Compression Technologies

A consistent trend in connection with digital audio transmission technologies has been continuous technological innovation leading to greater bandwidth optimization. Since the invention of digital audio, numerous new technologies have helped improve audio compression. Whether it was with perceptual audio coding technologies used in broadcast or consumer electronics applications such as MPEG-1 Layer-III (MP3), or later ones such as AAC or WMA, standard audio compression technologies have advanced greatly to allow near-CD quality sound with much lower bit rates.

In the past decade, with the rapid development of mobile networking technology, audio compression has experienced another major string of innovations, bringing voice, speech, and music compression to a whole new level. These advancements have brought forth a whole new generation of very low bit rate services.

Furthermore, with the greater ability to use increasingly cost-effective processing power in both encoding and decoding platforms, codec optimization and audio processing techniques are now widely available. Whether the solution serves as a codec pre-processing, "surround-over-stereo" (Dolby ProLogic, MPEG-Surround or Neural-THX Surround), or post-processing, the end result of each solution is to deliver more content at lower bit rates in order to increase the entertainment choices available to consumers.

3. Introduction to Neural Audio's Codec Optimization and Audio Pre-Processing

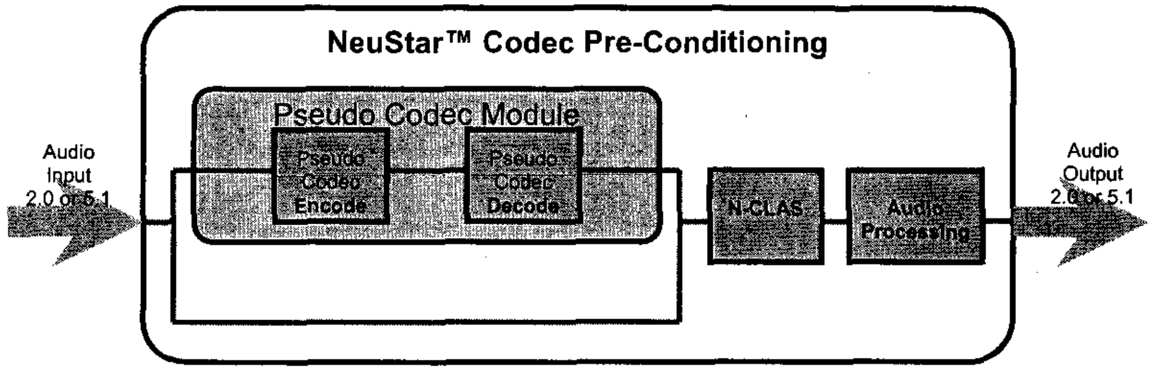
Neural Audio has long been involved in this process of technological innovation. Indeed, Neural Audio is a leader in codec optimization, audio processing and surround sound technologies. Working with partners such as Harris Corporation, VoiceAge Corporation and THX Ltd., Neural Audio has developed brands such as Neural-THX® Surround, Neural-AMR-WB+ and NeuStar™. Radio, TV, mobile networks and consumer electronics customers include XM as well as companies such as Classical Public Radio Network (CPRN), National Public Radio (NPR), ClearChannel, the BELO Group, Yamaha Corporation, Sony Corporation, Pioneer Electronics, Onkyo Corporation, Denon Electronics, and Marantz Electronics. Our business address is 11410 NE 122nd Way, Office 100, Kirkland, WA 98034.

Neural Audio introduced its codec optimization and audio pre-processing technology, commercially known as NeuStar™, in 2000. This solution provides radio broadcasters increased codec efficiency by optimizing bit rate compression and improving audio quality. In 2007, Neural Audio, in partnership with VoiceAge Corporation and other mobile technology partners, announced a new generation of its solution, fully customized and tuned for mobile operators and mobile service providers to deliver higher quality audio content over mobile networks. Neural Audio's solutions have won awards at several annual conventions of the National Association of Broadcasters.

The technology uses proprietary techniques, known as N-CLAS (Neural Codec Load Analysis System), which analyzes codec load to correct audio spectrum before actual encoding. The process dramatically reduces codec load (bit rate) while increasing final audio quality. Moreover, combined with advanced perceptual audio pre-processing techniques, Neural Audio is able to further increase audio quality at the consumer level. Figure 1 presents a description of the general processing.

Figure 1:

Description of Neural Audio's Codec Optimization and Audio Pre-Processing



Typical pre-processing techniques utilize a pseudo codec encode/decode system which simulates the compression effects of a given codec. A perceptual auditory model is applied to the simulated signal to identify the frequency bands of the audio which contain the most audible quantization artifacts when compressed by the given codec. Processing of the audio signal can be focused on those frequency bands with the highest levels of perceivable artifacts such that subsequent audio compression may result in lessened levels of audible distortions. Typical processing methods consist of attenuation or amplification of the energy of a given frequency band, and/or modifications to the coherence or phase of a given frequency band.

4. Codec Optimization and Audio Pre-Processing: Used By XM and Many Others

XM is among the companies that employ Neural Audio's solution. This technology, however, is not exclusive to satellite radio. Indeed, the core technology behind Neural Audio's Codec Optimization and Pre-Processing is codec agnostic, meaning that it can be used and tuned (optimized) for any audio codec, such as that used by Sirius. Neural Audio has worked with numerous industry leaders—in addition to XM—to adapt its solution to be used in various applications, tailored to the specifics of each codec used in each application.

Thus, whether for digital AM or FM Radio, HD Radio, Satellite Radio, Internet *Streaming or Mobile Streaming applications*, Neural Audio has been able to customize versions of its solution that optimize to each specific codec. As a result, numerous broadcasters are now able to improve audio quality and bit rate coding efficiency, and to make making multicasting a reality. Other entities use alternative technological methods to achieve the same goal, some of which are described in the separate technical report on compression technologies prepared by Dr. Deepen Sinha of ATC Labs.

Such technologies will continue to be available—and in fact, are likely to improve further—to a combined satellite radio provider, just as they will be available to any broadcaster. For example, a technology known as variable bit rate (VBR) compression has been developed over the past decade and has been widely adopted in telephony applications. Though not commonly used in radio broadcast up to now, this technology, often backward compatible with existing audio codecs, would allow for "multicasters" such as XM to know in real-time each channel's bit rate requirements and allocate bit rate as needed within the overall available bandwidth, thus making each channel as efficient as needed. Combined with other efficient automation programming solutions, VBR technology could further push the limit of bit rate efficiency and allow for more channels to be broadcasted over the same bandwidth. This is one example of the type of innovations that could be added to existing technological options and used to provide more programming and content within the same bandwidth resources.

5. Response to NAB Engineering Statement

The NAB Engineering Statement asserts (at page 2) that "[t]he data capacities of both the XM and Sirius systems are filled with programming and significant spare capacity is not available," and (at page 9) that "[b]oth providers are limited in their ability to add new program channels to their services without removing an equivalent number of existing program channels." These statements are flawed in at least five separate respects.

First, the conclusion that the satellite radio providers could only add new channels by subtracting others overlooks the various methods available today—including the solution developed by Neural Audio—that allow providers of audio entertainment to use their existing bandwidth more efficiently without any negative impact on audio quality. There is no evidence that either satellite radio provider has reached its "capacity" and can no longer avail itself of these technologies. In fact, XM has added three sports channels since the NAB Engineering Statement was issued last March.

Second, the nature of certain programming allows the companies to conserve bandwidth, even without resorting to the available technological techniques. For example, the existing systems offer seasonal sports such as the National

Football League (on Sirius) and Major League Baseball (on XM). While such programming requires a large bandwidth in order to support the many games being played simultaneously, this bandwidth allocation is largely unused much of the time. Since the minimum overlap exists between the NFL and MLB seasons, both sports packages could be offered to current XM and Sirius subscribers without regard to bandwidth constraints.

Third, in support of its claim that compression techniques would degrade audio quality, the NAB Engineering Statement (at page 6) cites "[e]xhaustive testing . . . by various organizations" to identify the minimum bit rate at which audio degradation might occur. The only test cited is one conducted by iBiquity on behalf of National Public Radio ("NPR") in 2004 involving the HDC codec. ("Perceptual Tests of iBiquity's HD Coder At Multiple Bit Rates," Prepared for National Public Radio by Ellyn G. Sheffield, Sheffield Audio Consulting, Oct. 14, 2004, *at* http://www.nprlabs.org/public/research/PerceptualMBR.pdf.) The NAB Engineering Statement describes this study as showing that "there is perceptible audio degradation at bit rates lower than about 48 kbps (kilobits per second)" and that "very good audio quality is achieved with bit rates around 64-96 kbps."

That NPR test, however, does not provide any indication of the bit rates that would be needed on XM's system to maintain audio quality. The premise of this argument is that the HDC codec tested in the NPR study is "similar" (see page 6) to the one employed by XM. However, that is not true. As an initial matter, the NAB Engineering Statement incorrectly identifies XM's codec as MPEG-AAC. In fact, XM's audio codec utilizes a proprietary version of the aacplus (MPEG 4 HE AAC) open standard, which uses Spectral Band Replication (SBR) to further enhance compression efficiency. Further, as explained by the President and Chief Executive Officer of iBiquity—which, as noted, conducted the NPR study on which the NAB Engineering Statement relies for this point—the HDC codec "is not aacPlus." (Leslie Stimson, *HD Radio: Will New Codec Do the Trick?*, RADIO WORLD, Sept. 10, 2003, *at* http://www.radioworld.com/reference-room/iboc/01_rw_hd_codec_2-09.10.03.shtml.) Rather, the HDC Codec has been customized for AM and FM broadcasters and to work with the iBiquity system.

Even apart from the fact that it addressed a different codec, the NPR test did not involve the Codec Optimization and Audio Pre-Processing technology described above, which XM uses to improve audio quality. Finally, a more recent report prepared for NPR by iBiquity shows that a new alternative to the audio codec used in HD Radio, VoiceAge's AMR-WB+, would allow bandwidth-limited broadcasters to be more efficient. ("Report On Perceptual Tests of Coders at Low- and Very Low-Bit Rates," Prepared for National Public Radio and IAAIS In Cooperation with iBiquity Digital Corporation, by Ellyn G. Sheffield, Sheffield Audio Consulting, *at* www.npr.org/euonline/pub/iboc/low_bit_rate_coder_report.pdf.) In short, the

NPR study provides no support for the claim that XM (or, for that matter, Sirius) are unable to achieve increased bandwidth efficiency.

Fourth, the NAB Engineering Statement misstates the alleged channel capacity of both companies. The NAB Engineering Statement (at page 6) states that XM and Sirius offer 148 and 123 channels, respectively, which it then claims represent the maximum number of channels that each company can provide. In fact, these figures significantly *understate* the number of channels provided by each company: XM now offers 178 channels while Sirius offers 134. Thus, XM and Sirius already offer many channels above the limit at which the NAB Engineering Statement asserts audio degradation will occur. This fact alone undermines the validity of the bit-rate analysis contained in the NAB Engineering Statement.

Finally, the claim in the NAB Engineering Statement (at page 5) that "the total capacity of the two systems will not change even if the companies merge" is incorrect. We understand that XM and Sirius are in stages of expanding system bandwidth by up to 25 percent of their total system capacity through the introduction of hierarchical modulation technology. In hierarchical modulation, two separate data streams are modulated on a single carrier. The hierarchical modulation consists of a basic constellation (modulation scheme), which is the same as in the original system, and a secondary constellation (overlaid on top of the original), which carries the additional data for the upgraded system. The upgraded system with the hierarchical modulation is backward compatible in the sense that receivers that have been deployed in the original system can continue receiving data in the basic constellation. New receivers can be designed to receive data carried in the secondary constellation, as well as those in the basic constellation. As a merged company, this new bandwidth could be pooled to support expanded programming for subscribers.

6. Conclusion

In sum, (1) numerous technologies allow broadcasters to distribute more content at the same perceived audio quality, (2) Neural Audio's Codec Optimization and Audio Pre-Processing technology tuned for XM's codec already has delivered numerous channels and other improvements without sacrificing audio quality, and (3) new technologies used in other applications can easily be applied to satellite radio broadcasts to further improve bit rate efficiency and audio quality. Based on these factors, Neural Audio believes that the claims described in the NAB Engineering Statement do not represent any major limitations preventing XM and Sirius from offering an increased number of channels to their customers while preserving audio quality.